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Objective

A research career in a well-established institution, which allows for adequate time and funding directed towards advancing the state-of-the-art.

Interests

My research interests include signal processing, machine learning, deep learning, robust speech recognition and audio applications like recognition, enhancement and coding.

Current Position

I am an Assistant Professor at the Electrical Engineering, Indian Institute of Science, Bangalore. I manage the Learning and Extraction of Acoustic Patterns (LEAP) laboratory where the activities are focussed on information extraction and analysis of acoustic and acoustic like signals.

Education

PhD, Center of Language and Speech Processing (4.0/4.0) — Jan. 2009-Dec. 2011
Johns Hopkins University, Baltimore, USA.

Master of Engineering, Signal Processing (7.4/8.0) — Aug. 2004-July. 2006
Indian Institute of Science, Bangalore, India.

Bachelor of Technology, Electronics and Communications (82%) — Oct. 2000- June. 2004
College of Engineering, Trivandrum, India

Skills

Programming: MATLAB, C, Python, HTML.

Tools: HTK, QuickNet, Latex.

Operating Systems: Unix/Linux, Windows, Mac OS X

Honors and Awards

Best outgoing student award for M.E. Signal Processing, Dept. of ECE, Indian Institute of Science, Bangalore, 2006.

High Quality Tutorial Presentation on “The Art and Science of Speech Feature Engineering” in Interspeech, 2014, Singapore.

Experience

Research Staff Member at IBM T.J Watson Center *Dec.2011- Dec. 2015*

Reasearch on signal analysis and processing of noisy and degraded radio channel speech for biometric applications like speaker and language recognition as well as speech activity detection. These technologies are developed for the U.S. Government under the Defense Advanced Research Project Agency. Full-time employment (40 hours per week).

Research Intern at IBM T.J Watson Center *June 2010-Aug 2010*

The main focus of this internship was to develop feature normalization techniques for speaker verification in far-field reverberant environments. Full-time employment (40 hours per week).

Research Assistant in Idiap Research Institute, Switzerland *Oct. 2006 - Jan 2009*

The goal of this work was to investigate the use of long-term energy summarization of speech signals for speech recognition and coding applications. Full-time employment (40 hours per week).

External Talks and Seminars

“Deep Learning for Speech Processing”, IBM India Research Labs, February 2016.

“The Art and Science of Speech Feature Engineering”, Tutorial at Interspeech, Singapore, Sept. 2014.

“Algorithms in Speech Signal Processing”, College of Engineering, Trivandrum, India, Nov., 2013.

“Robust Processing of Noisy and Degraded Channel Speech’’, Computational Science an Artificial Intelligence Laboratory, MIT, Cambridge, USA, Oct., 2013.

“Dealing with Noisy Speech Using Autoregressive Models’’, Idiap Research Institute, Martigny, Switzerland, Aug, 2013.

“Signal analysis using autoregressive models of amplitude modulation’’, Electrical and Computer Engineering, University of Texas, Dallas, Feb. 2013.

‘’Frequency Domain Linear Predictive Analysis of Speech’’, Indian Institute of Science, Bangalore, India, Nov. 2011.

‘’Signal modeling with long term feature processing’’, Raytheon BBN Technologies, Cambridge, MA, USA, July 2011.

Professional Memberships

IEEE Signal Processing Society Member

International Speech Communication Association (ISCA)

Teaching

Course Instructor for “E9-261 - Speech Information Processing”, Indian Institute of Science, Jan-April, 2016. (Jointly given with Dr. Prasanta Ghosh).

Teaching Assistant for ‘’Information processing of sensory signals’’, for Spring 2009, Spring 2010, Johns Hopkins University.

Patents

“Method for System Combination in Audio Analytics Application”, with IBM Watson Center, USA. [Filed June 2015].

“Spectral Noise Shaping in Audio Coding Based on Spectral Dynamics in Frequency Sub-bands”, with Qualcomm Inc, [Approved Nov. 2011].

“Temporal Masking in Audio Coding Based on Spectral Dynamics in Frequency Subbands”, with Qualcomm Inc, [Approved May 2009].

Publications

I. Thesis

S. Ganapathy, "Signal Analysis using Autoregressive Models of Amplitude Modulation ", Johns Hopkins University, Jan. 2012

II. Peer Reviewed Journals

S. Ganapathy, M. Omar, “Auditory Motivated Front-end for Noisy Speech Using Spectro-temporal Modulation Filtering”, Journal of Acoustical Society of America, EL343-349, Vol. 136(5), Nov. 2014.

S. Ganapathy, S. H. Mallidi and H. Hermansky, "Robust Feature Extraction Using Modulation Filtering of Autoregressive Models", IEEE Transactions on Audio, Speech and Language Processing, Vol. 22(8), pp. 1285-1295, Aug. 2014.

S. Ganapathy and J. Pelecanos, "Enhancing Frequency Shifted Speech Signals in Single Side Band Communication", IEEE Signal Processing Letters, Vol. 20(12), pp. 1231-1234, Oct. 2013.

S. Ganapathy and H. Hermansky, "Temporal Resolution Analysis in Frequency Domain Linear Prediction", Journal of Acoustical Society of America, EL436-442, Vol. 132(5), Oct. 2012.

S. Ganapathy, S. Thomas and H. Hermansky, "Temporal envelope compensation for robust phoneme recognition using modulation spectrum", Journal of Acoustical Society of America, Vol. 128(6), pp. 3769-3780, Dec. 2010.

S. Ganapathy, P. Motlicek and H. Hermansky, "Autoregressive Models Of Amplitude Modulations In Audio Compression", IEEE Transactions on Audio, Speech and Language Processing, Vol. 18(6), pp.1624-1631, Aug. 2010.

P. Motlicek, **S. Ganapathy**, H. Hermansky and H. Garudadri, "Wide-Band Audio Coding based on Frequency Domain Linear Prediction", EURASIP Journal on Audio, Speech, and Music Processing, Vol. 2010(3), pp. 1-14, Jan. 2010.

S. Ganapathy, S. Thomas and H. Hermansky, "Modulation Frequency Features For Phoneme Recognition In Noisy Speech", Journal of Acoustical Society of America, EL8-12, Vol. 125(1), Jan. 2009.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Recognition of Reverberant Speech Using Frequency Domain Linear Prediction", IEEE Signal Processing Letters, Vol. 15, pp. 681-684 Nov. 2008.

III. Conferences

S. Sadjadi, **S. Ganapathy** and J. Pelecanos, "The IBM 2016 Speaker Recognition System", Odyssey, Spain, June, 2016.

S. Sadjadi, **S. Ganapathy** and J. Pelecanos, “Speaker Age Estimation On Conversational Telephone Speech Using Senone Posterior Based I-vectors”, ICASSP, Shanghai, 2016.

S. Ganapathy, S. Thomas, D. Dimitriadis, S. Rennie "Investigating Factor Analysis Features for Deep Neural Networks In Noisy Speech Recognition", Interspeech, Dresden, Germany, Sept. 2015.

S. Ganapathy, "Robust Speech Processing Using ARMA Spectrograms", ICASSP, Brisbane, April, 2015.

S. Sadjadi, J. Pelecanos and **S. Ganapathy**, "Nearest Neighbor Discriminant Analysis for Language Recognition", ICASSP, Brisbane, April, 2015.

S. Ganapathy, K. J. Han, M. Omar, M. V. Segbroeck and S. Narayan, "Robust Language Identification Using Convolutional Neural Networks", Interspeech, Singapore, 2014.

S. Thomas, **S. Ganapathy**, G. Saon and H. Soltau, “Analyzing Convolutional Neural Networks For Speech Activity Detection In Mismatched Acoustic Conditions”, ICASSP, Florence, 2014.

M. Omar and **S. Ganapathy**, “Shift-Invariant Features for Speech Activity Detection in Adverse Radio-Frequency Channel Conditions”, ICASSP, Florence, 2014.

G. Saon, S. Thomas, H. Soltau, **S. Ganapathy** and B. Kingsbury, "The IBM Speech Activity Detection System for the DARPA RATS Program", Interspeech, Lyon, Aug. 2013.

K. J. Han, **S. Ganapathy**, M Li, M. Omar and S. Narayan, "TRAP Language Identification System for RATS Phase II Evaluation", Interspeech, Lyon, Aug. 2013.

H. Mallidi, **S. Ganapathy** and H. Hermansky, "Robust Speaker Recognition Using Spectro-Temporal Autoregressive Models", Interspeech, Lyon, Aug. 2013.

S. Ganapathy, M. Omar and J. Pelecanos, "Unsupervised Channel Adaptation For Language Identification Using Co-training", ICASSP, Vancouver, May, 2013.

S. Ganapathy, M. Omar and J. Pelecanos, "Noisy Channel Adaptation in Language Identification", IEEE SLT, Miami, Dec, 2012.

S. Ganapathy and H. Hermansky, "Robust Phoneme Recognition Using High Resolution Temporal Envelopes", Interspeech, Portland, Sept. 2012.

S. Thomas, **S. Ganapathy**, A. Jansen and H. Hermansky, "Data-driven Posterior Features for Low Resource Speech Recognition Applications", Interspeech, Portland, Sept. 2012.

S. Ganapathy, S. Thomas and H. Hermansky, "Feature Extraction Using 2-D Autoregressive Models For Speaker Recognition", ISCA Speaker Odyssey, June 2012.

S. Thomas, H. Mallidi, **S. Ganapathy** and H. Hermansky, "Adaptation Transforms of Auto-Associative Neural Networks as Features for Speaker Verification", ISCA Speaker Odyssey, June 2012.

D. Gomero et al. "The UMD-JHU 2011 Speaker Recognition System", ICASSP, Japan, Mar. 2012.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Multilingual MLP Features For Low-resource LVCSR Systems", ICASSP, Japan, Mar. 2012.

S. Ganapathy, P. Rajan and H. Hermansky, "Multi-layer Perceptron Based Speech Activity Detection for Speaker Verification", IEEE WASPAA, Oct. 2011.

H. Mallidi, **S. Ganapathy** and H. Hermansky, "Modulation spectrum analysis for recognition of reverberant speech", Interspeech, Italy, Aug. 2011.

S. Ganapathy, J. Pelecanos and M. Omar, "Feature Normalization for Speaker Verification in Room Reverberation", ICASSP, Prague, May 2011.

S. Garimella, **S. Ganapathy** and H. Hermansky, "Sparse Auto-associative Neural Networks: Theory and Application to Speech Recognition", Interspeech, Japan, Sept. 2010.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Cross-lingual and Multi-stream Posterior Features for Low-resource LVCSR Systems", Proc. of Interspeech, Japan, Sept. 2010.

S. Thomas, K. Patil, **S. Ganapathy**, N. Mesgarani, H. Hermansky, "A Phoneme Recognition Framework based on Auditory Spectro-Temporal Receptive Fields", Proc. of Interspeech, Japan, Sept. 2010.

S. Ganapathy, S. Thomas and H. Hermansky, "Robust Spectro-Temporal Features Based on Autoregressive Models of Hilbert Envelopes", ICASSP, Dallas, USA, March 2010.

S. Ganapathy, S. Thomas and H. Hermansky, "Comparison of Modulation Features For Phoneme Recognition", ICASSP, Dallas, USA, March 2010.

S. Ganapathy, S. Thomas, and H. Hermansky, "Temporal Envelope Subtraction for Robust Speech Recognition Using Modulation Spectrum", IEEE ASRU, 2009.

S. Ganapathy, S. Thomas, P. Motlicek and H. Hermansky, "Applications of Signal Analysis Using Autoregressive Models for Amplitude Modulation", IEEE WASPAA 2009.

S. Ganapathy, S. Thomas and H. Hermansky, "Static and Dynamic Modulation Spectrum for Speech Recognition", Proc. of Interspeech, Brighton, UK, Sept. 2009.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Tandem Representations of Spectral Envelope and Modulation Frequency Features for ASR", Proc. of Interspeech, Brighton, UK, Sept. 2009.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Phoneme Recognition Using Spectral Envelope and Modulation Frequency Features", ICASSP, Taiwan, April 2009.

S. Ganapathy, S. Thomas and H. Hermansky, "Front-end for Far-field Speech Recognition based on Frequency Domain Linear Prediction", Proc. of INTERSPEECH, Brisbane, Australia, Sep 2008.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Hilbert Envelope Based Specto-Temporal Features for Phoneme Recognition in Telephone Speech", Proc. of INTERSPEECH, Brisbane, Australia, Sep 2008.

S. Ganapathy, P. Motlicek, H. Hermansky and H. Garudadri, "Spectral Noise Shaping: Improvements in Speech/Audio Codec Based on Linear Prediction in Spectral Domain", Proc. of INTERSPEECH, Brisbane, Australia, Sep 2008.

P. Motlicek, **S. Ganapathy**, H. Hermansky, H. Garudadri and Marios Athineos, "Perceptually motivated Sub-band Decomposition for FDLP Audio Coding", in Lecture Notes In Artificial Intelligence, Springer-Verlag Berlin, Heidelberg, 2008.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Spectro-Temporal Features for Automatic Speech Recognition using Linear Prediction in Spectral Domain", Proc. of EUSIPCO, Lausanne, Switzerland, Aug 2008.

S. Ganapathy, P. Motlicek, H. Hermansky and H. Garudadri, "Autoregressive Modelling of Hilbert Envelopes for Wide-band Audio Coding", AES 124th Convention, Audio Engineering Society, May 2008.

S. Ganapathy, P. Motlicek, H. Hermansky and H. Garudadri, ""Temporal Masking for Bit-rate Reduction in Audio Codec Based on Frequency Domain Linear Prediction", Proc. of ICASSP, April 2008.

S. Thomas, **S. Ganapathy** and H. Hermansky, "Hilbert Envelope Based Features for Far-Field Speech Recognition", Lecture Notes in Computer Science, Springer Berlin, Heidelberg 2008.

P. Motlicek, H. Hermansky, **S. Ganapathy** and H. Garudadri, "Frequency Domain Linear Prediction for QMF Sub-bands and Applications to Audio Coding", Lecture Notes in Computer Science, Springer Berlin, Heidelberg 2007.

P. Motlicek, H. Hermansky, **S. Ganapathy** and H. Garudadri, "Non- Uniform Speech/Audio Coding Exploiting Predictability of Temporal Evolution of Spectral Envelopes", Lecture Notes in Computer Science, Springer Berlin, Heidelberg 2007.